PATENT ABSTRACTS OF JAPAN

(11)Publication number : 07-162985

(43)Date of publication of application : 23.06.1995

(51)Int.Cl. H04R 3/00
G10K 15/00
H04R 3/04
H04R 5/02
H04S 5/02

(21)Application number : 05-302910

(22)Date of filing : 02.12.1993

(71)Applicant : FUJITSU TEN LTD

(72)Inventor : YAMAGUCHI HIROYUKI
KOWAKI HIROSHI

(54) DEVICE FOR AUTOMATICALLY CORRECTING LISTENING POSITION

(57)Abstract:

PURPOSE: To eliminate the phase interference in all bands by providing the device for automatically correcting listening position capable of automatically and easily correcting the difference in distance, that is, the time difference from each speaker unit in each speaker system to the listening position in the on-vehicle digital audio system.

CONSTITUTION: The signal of each channel to each speaker system at the time of reproduction is divided in band by filters 21 and 22 having the cut-off frequency the same as the crossover frequency of the corresponding speaker system. The divided signal is delayed by a delay device 23 by the delay value for correcting the speaker unit obtained by the measurement in addition. The delayed signal for each speaker unit is mixed by a mixer 24 into the signal of each corresponding speaker system and then it is outputted to a split circuit 75 at the speaker system side.

LEGAL STATUS

[Date of request for examination]	26.10.2000
[Date of sending the examiner's decision of rejection]	
[Kind of final disposal of application other than the examiner's decision of rejection or application converted registration]	
[Date of final disposal for application]	
[Patent number]	3148060
[Date of registration]	12.01.2001
[Number of appeal against examiner's decision of rejection]	
[Date of requesting appeal against examiner's decision of rejection]	
[Date of extinction of right]	

CLAIMS

[Claim(s)]
[Claim 1] It is equipment for amending automatically in the digital audio system for mount, the range difference, i.e., the time difference, from each loudspeaker unit in each loudspeaker system to a listening location. About said each loudspeaker unit, by M sequence signal regeneration means to reproduce an M sequence signal, and sound-collecting, with the predetermined microphone put on said listening location An M sequence response acquisition means to acquire the M sequence response to said M sequence signal, and to store in predetermined memory, An impulse response restoration means to restore said M sequence response to an impulse response by high-speed M sequence conversion, A threshold setting means to search the peak value of said impulse response, to start based on this peak value, and to set up the threshold for detection, A build-up-time detection means to detect the build up time which is the time amount which exceeds said threshold first, It is based on the loudspeaker unit which has the longest build up time in said each loudspeaker unit. the difference of the build up time of each of said loudspeaker unit -- asking -- this -- each -- with a delay value calculation means to compute difference as a delay value at the time of playback of each of said loudspeaker unit The signal band division means which carries out band division with the filter which has a cut off frequency equivalent to the crossed frequency of the loudspeaker system which corresponds the signal of each channel to said each loudspeaker system at the time of playback, A signal delay means to make it only the delay value of the corresponding loudspeaker

unit asked for each signal divided by said signal band division means by said delay value calculation means delayed, The listening location automatic compensator which comes to provide a signal mixing means to output to the dividing network by the side of said each corresponding loudspeaker system after mixing the signal for every [you were made to be delayed with said signal delay means] loudspeaker unit to the signal of each corresponding loudspeaker system.
[Claim 2] The listening location automatic compensator according to claim 1 which computes the average of the direct-current-offset level in the fixed section before the standup of said impulse response, and possesses further the direct-current-offset removal means which subtracts this average from said impulse response before the setup of the threshold by said threshold setting means.

DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Industrial Application] This invention relates to the listening location automatic compensator which is equipment for amending automatically the bias of the sound field produced from each loudspeaker according to the difference of each distance to a listening location in the digital audio system for mount.
[0002]
[Description of the Prior Art] Conventionally, such a listening location automatic compensator is indicated by the specification attached to Japanese Patent Application No. No. 219842 [five to] which is the patent application by the applicant for this patent. The listening location automatic compensator indicated by it attains automatic amendment by searching for the time difference by each loudspeaker by making an M sequence (maximum periodicity train) signal strong against a noise into a measurement signal by one of the pseudo-random signals, and detecting the build up time of a signal from the restored impulse response, and setting up the time subtraction as delay of each loudspeaker at the time of playback. And when the configuration of a loudspeaker is multiway, performing time difference amendment for one loudspeaker unit (band loudspeaker) which reproduces the large band (500Hz - 2kHz) of the effect to the normal position is proposed.
[0003]
[Problem(s) to be Solved by the Invention] When the configuration of a loudspeaker of such a listening location automatic compensator concerning the conventional technique was one way, there was especially no problem, but since a setup of the delay optimal about loudspeaker units other than the loudspeaker unit set as the

object of time difference amendment as mentioned above in the case of multiway will be performed, it has the problem of producing interference of a phase.

[0004] Therefore, also in the case of a multiway loudspeaker system, time difference is measured for every loudspeaker unit, and although it is necessary to improve so that it may amend automatically, the following problems occur about a setup of the delay at the time of playback in that case. That is, drawing 5 is a block diagram which illustrates the configuration concerning the conventional technique for setting up time delay to each loudspeaker unit. In this drawing, a sign 20 shows DSP (digital signal processor) used as the center of a digital audio system. Moreover, Signs 70a, 70b, 70c, and 70d show the loudspeaker system front right-hand side (FR), front left-hand side (floor line), rear right-hand side (RR), and on the left-hand side of rear (RL) one, and Signs 71a, 71b, 71c, and 71d show the loudspeaker unit of Ufa (woofer), and they show the loudspeaker unit of a tweeter (loudspeaker for loud sounds) 72a, 72b, 72c, and 72d, respectively. Furthermore, a sign 74 shows the crossover network (it is also called a channel divider, a channel filter, or a dividing network.) of a digital method, in the sign 21 of the interior, a high pass filter and a sign 23 show delay (delay machine), and, as for a low pass filter and a sign 22, a sign 25 shows a D/A converter.

[0005] And although the software of DSP20 of a digital audio system measures the time delay for amendment about each loudspeaker unit in such a configuration, the time delay can be given and the software of a crossover network 74 carries out delay processing by delay 23, since the means which hands over time delay data from DSP20 to a crossover network 74 does not have anything, it is in a very difficult situation to realize automatic amendment processing. Moreover, this crossover network is also very expensive.

[0006] In view of this actual condition, the purpose of this invention is in the digital audio system for mount by offering the advanced listening location automatic compensator which can amend automatically and easily, the range difference, i.e., the time difference, from each loudspeaker unit in each loudspeaker system to a listening location, to lose phase interference in all bands.

[0007]

[Means for Solving the Problem] This invention carries out band division of the audio signal with the filter which has a cut off frequency equivalent to the crossed frequency of a loudspeaker, and after it gives and mixes delay (mixing), it attains the above-mentioned purpose paying attention to outputting to a loudspeaker side. Namely, the listening location automatic compensator concerning this invention It is equipment for amending automatically in the digital audio system for mount, the range difference, i.e., the time difference, from each loudspeaker unit in each loudspeaker system to a listening location. About said each loudspeaker unit, by M sequence signal regeneration means to reproduce an M sequence signal, and sound-collecting, with the predetermined microphone put on said listening location An M sequence response acquisition means to acquire the M sequence response to said M sequence signal, and

to store in predetermined memory, An impulse response restoration means to restore said M sequence response to an impulse response by high-speed M sequence conversion, A threshold setting means to search the peak value of said impulse response, to start based on this peak value, and to set up the threshold for detection, A build-up-time detection means to detect the build up time which is the time amount which exceeds said threshold first, It is based on the loudspeaker unit which has the longest build up time in said each loudspeaker unit. the difference of the build up time of each of said loudspeaker unit -- asking -- this -- each -- with a delay value calculation means to compute difference as a delay value at the time of playback of each of said loudspeaker unit The signal band division means which carries out band division with the filter which has a cut off frequency equivalent to the crossed frequency of the loudspeaker system which corresponds the signal of each channel to said each loudspeaker system at the time of playback, A signal delay means to make it only the delay value of the corresponding loudspeaker unit asked for each signal divided by said signal band division means by said delay value calculation means delayed, After mixing the signal for every [you were made to be delayed with said signal delay means] loudspeaker unit to the signal of each corresponding loudspeaker system, it is the listening location automatic compensator which comes to provide a signal mixing means to output to the dividing network by the side of said each corresponding loudspeaker system.
[0008] Moreover, according to this invention, further, said listening location automatic compensator computes the average of the direct-current-offset level in the fixed section before the standup of said impulse response, and possesses the direct-current-offset removal means which subtracts this average from said impulse response before the setup of the threshold by said threshold setting means.
[0009]
[Function] In the above-mentioned listening location automatic compensator, about each loudspeaker unit, an M sequence signal is reproduced, the response is acquired, and it is stored in predetermined memory. And the M sequence response is restored to an impulse response by high-speed M sequence conversion. And the peak value of the impulse response is searched, it starts based on the peak value, and the threshold for detection is set up. And the build up time which is the time amount which exceeds the threshold first is detected. and the difference of the build up time of each loudspeaker unit asks in each loudspeaker unit on the basis of the loudspeaker unit which has the longest build up time -- having -- the -- each -- let difference be the delay value which should be set up at the time of playback of each loudspeaker unit. And the signal of each channel to each loudspeaker system at the time of playback Band division is carried out with the filter which has a cut off frequency equivalent to the crossed frequency of a corresponding loudspeaker system, and each divided signal You are made only for the delay value of the corresponding loudspeaker unit called for previously to be delayed, and the signal for every [you were made to be

delayed] loudspeaker unit is outputted to the dividing network by the side of each loudspeaker system, after being mixed to the signal of each corresponding loudspeaker system. In this way, measurement of time difference and a setup of the correction value by it are performed automatically [in the same digital signal processor] and easily. Moreover, if it has the aforementioned direct-current-offset removal means further, the direct current offset which may ride on an impulse response will be removed.

[0010]

[Example] Hereafter, the example of this invention is explained with reference to drawing 1 of an accompanying drawing - drawing 4 .

[0011] Drawing 2 is the outline block diagram showing the hardware configuration at the time of time difference measurement of the listening location automatic compensator concerning one example of this invention. In this drawing a microcomputer (microcomputer) and a sign 20 a sign 10 A digital signal processor (DSP), Digital one / analog (D/A) transducer, and a sign 40 a sign 30 An analog / digital (A/D) transducer, A sign 50 microphone amplifier and sign 70a for loudspeaker amplifier and a sign 60 FR loudspeaker system, sign 70b -- in floor line loudspeaker system and sign 70c, a right ear microphone and a sign 82 show a left ear microphone, and, as for RR loudspeaker system and 70d of signs, a sign 90 shows sound field, as for RL loudspeaker system and a sign 81. In addition, each loudspeaker systems 70a, 70b, 70c, and 70d have Ufa, and each loudspeaker unit and dividing network of a tweeter, respectively. Moreover, microphones 81 and 82 are put on a listening location.

[0012] Next, in the configuration of drawing 2 , a microcomputer 10 and DSP20 explain the procedure of measurement of the time difference by the listening location collaborated and performed based on the flow chart of drawing 3 . First, about one of the loudspeaker units, an M sequence signal is reproduced and the response is stored in predetermined memory (step 102). However, when the loudspeaker unit of right-hand side loudspeaker system 70a or 70c is reproduced and left-hand side loudspeaker system 70b or a 70d loudspeaker unit is reproduced with the right ear microphone 81, a response is acquired with the left ear microphone 82.

[0013] Subsequently, the calculated response of an M sequence is restored to an impulse response which is illustrated by drawing 4 using high-speed M sequence conversion (step 104). Since high-speed M sequence conversion is a well-known thing, it omits explanation especially here. Next, the noise average of the fixed section before signal arrival (it illustrates to drawing 4) is calculated, and it is set up as direct current offset (step 106). Subsequently, the direct current offset is subtracted from an impulse response (step 108). Thus, before starting and setting up the threshold for detection, a standup can be detected more to accuracy by removing direct current offset.

[0014] And as shown in drawing 4 , the peak of an impulse response is searched and it is based on the peak value, and it is threshold = peak value xk. (k is the constant of

arbitration)
By being alike and performing the shown operation, it starts and the threshold for detection is set up (step 110). Subsequently, the time amount which exceeds first the threshold for which it asked is detected as build up time (refer to drawing 4), i.e., time of concentration, (step 112).
[0015] The above processing is carried out about each loudspeaker unit, as shown in step 114. and criteria [loudspeaker unit / which starts most / late] -- carrying out -- the difference of the build up time of each other loudspeaker units -- asking -- the -- each -- difference is determined as the delay value which should be set up at the time of playback of each loudspeaker unit (step 116). In this way, by setting up the calculated delay value to each loudspeaker unit at the time of the usual playback, amendment of the bias of the sound field by the listening location is attained. The delay setting processing is explained below.
[0016] Drawing 1 is a block diagram for explaining the processing configuration concerning this invention for setting up time delay to each loudspeaker unit. The same sign is given to the same element as drawing 5 , and if a different element is explained, the channel divider 75 of D/A converter 30 and a cheap analog form will be used instead of the crossover network 74 of drawing 5 . As shown in this drawing, in DSP20, band division of the signal of the channel to FR loudspeaker system 70a at the time of playback is carried out with the low pass filter 21 and high pass filter 22 which have a cut off frequency equivalent to the crossed frequency of loudspeaker system 70a. And it is made only for the delay value of the corresponding loudspeaker units 71a and 72a called for previously to be delayed with the delay vessel 23 to each divided signal. Subsequently, the signal for every [you were made to be delayed] loudspeaker unit is mixed by the mixer 24. And the signal is transmitted to the dividing network 75 by the side of loudspeaker system 70a through D/A converter 30. That is, amendment processing is performed when a loudspeaker side is reached. The completely same processing is made also to loudspeaker systems 70b, 70c, and 70d. In this way, measurement of time difference and a setup of the correction value by it are performed automatically [in the same digital signal processor 20] and easily.
[0017] As mentioned above, although one example of this invention has been described, it will be easy for this contractor for this invention not to be limited to this and to think out various examples, of course.
[0018]
[Effect of the Invention] As explained above, according to the listening location automatic compensator concerning this invention, turbulence of the bias of the sound field produced according to the range difference between each loudspeaker unit and a listening location and the frequency characteristics by phase interference crosses to all bands, and is amended. Moreover, since the M sequence signal is used at the time of measurement, it is hard to be influenced of a noise and exact amendment can be realized. Moreover, since it is realizable as DSP in a digital audio system, and a

program of a microcomputer, this equipment becomes unnecessary [a crossover network]. Furthermore, it is possible by removing offset to be able to detect a standup still more correctly and to make an amendment error small.

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]

[Drawing 1] It is a block diagram for explaining the processing configuration concerning this invention for setting up time delay to each loudspeaker unit.

[Drawing 2] It is the outline block diagram showing the hardware configuration at the time of time difference measurement of the listening location automatic compensator concerning one example of this invention.

[Drawing 3] It is the outline flowchart which shows the procedure of signal processing in the listening location automatic compensator concerning one example of this invention.

[Drawing 4] It is the timing diagram which shows the example of a measurement impulse response.

[Drawing 5] It is the block diagram which illustrates the configuration concerning the conventional technique for setting up time delay to each loudspeaker unit.

[Description of Notations]

10 -- Microcomputer
20 -- Digital signal processor
21 -- Low pass filter
22 -- High pass filter
23 -- Delay machine (delay)
24 -- Mixer
30 -- Digital to analog converter
40 -- An analog / digital converter
50 -- Loudspeaker amplifier
60 -- Microphone amplifier
70a -- Front right-hand side (FR) loudspeaker system
70b -- Front left-hand side (floor line) loudspeaker system
70c -- Rear right-hand side (RR) loudspeaker system
70d -- Rear left-hand side (RL) loudspeaker system
71a, 71b, 71c, 71d -- Ufa (loudspeaker unit)
72a, 72b, 72c, 72d -- Tweeter (loudspeaker unit)
74 -- Crossover network
75 -- Channel divider of an analog form

81 -- Right ear microphone
82 -- Left ear microphone
90 -- Sound field